Attachment III
NAS 9-11743

(NASA-CR-115631) ASYSTD LIBRARY: GROUP IDENTIFIERS GAUSSIAN NOISE GENERATOR (Systems Associates, Inc.) Apr. 1972 77 p

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78

GROUP IDENTIFIERS

SIGNAL GENERATORS

MODULATORS

DEMODULATORS

FILTERS

LIMITERS

TRANSFORMS

CONVERTERS

CODERS

MATH

MISCELLANEOUS

SYSTEMS
ASSOCIATES

MODEL	GROUP ID	PAGE	DATE
GAUSSIAN NOISE GENERATOR	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

GNOISE

DESCRIPTION

This function provides noise modeling capability, providing the SNR and ENB of the generator are defined.

USAGE

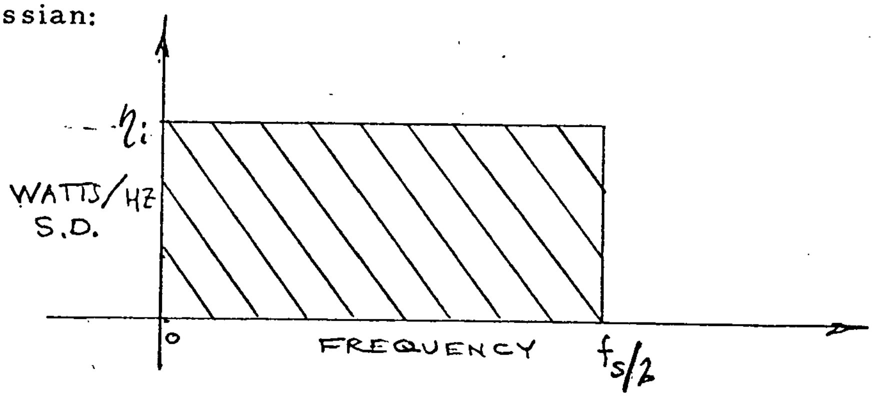
N1 < GNØISE (SNR, ENB, ISTART) > N2

where: SNR is the signal-to-noise ratio desired in ENB (equivalent noise bandwidth) assuming a 1 watt signal level.

ISTART is a positive integer (>0) for initializing the random number generator.

DETAILED DESCRIPTION

White Gaussian:



where

$$f_s = \frac{1}{DT} = \frac{1}{\text{sampling rate}}$$

$$\sigma_i^2 = \frac{\eta_i f_s}{2} = \frac{\eta_i}{2DT}$$

or

$$\boxed{\eta_i = 2\sigma_i^2 DT} \quad \text{(Watts/Hz)}$$

$$N_o = \eta_i * ENB = 2\sigma^2 DT * ENB \quad \text{(watts)}$$

where ENB = equivalent noise bandwidth under consideration.

For a given SNR in bandwidth, BW:

$$\frac{S}{N_o} = 10^{SNR/10}$$

where

$$S = \text{signal power in BW} \quad \text{(watts)}$$

or

$$N_o = S * 10^{-SNR/10} = 2\sigma_i^2 DT * ENB$$

or

$$\boxed{\sigma_i = \sqrt{S} / \sqrt{10^{SNR/10} * 2DT * ENB}}$$

APPLICATION

This model is a function and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
GAUSSIAN NOISE TWO	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

GNOIS2

DESCRIPTION

This model provides noise modeling capability, providing the spectral density desired.

USAGE

N1 <GNØIS2 (ETA, ISTART)> N2

where: ETA is the desired spectral density (watts/Hz)

ISTART is a positive integer (>0) for initializing the random number generator

DETAILED DESCRIPTION

See "Gaussian Noise Generator"

$$ETA = \frac{1}{10^{SNR/10} * ENB}$$

APPLICATION

This model is a function and may be used in expressions.

SYSTEMS ASSOCIATES

MODEL	GROUP ID	PAGE	DATE
PERIODIC TABLE FUNCTION	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

PERIODIC TABLE FUNCTION
PTABLE

DESCRIPTION

This model provides the periodic function capability. The output is periodic with period T5.

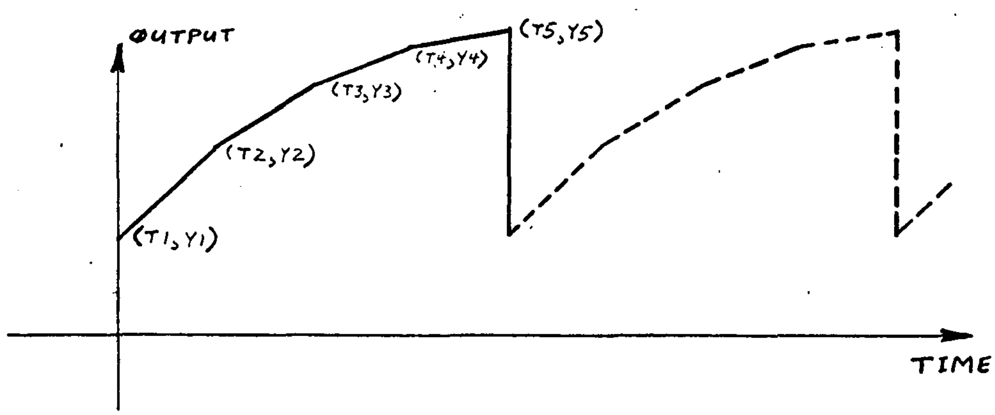
USAGE

N1 < PTABLE (T1, Y1, T2, Y2, T3, Y3, T4, Y4, T5, Y5) > N2

where: T1, Y1 ⋮ T5, Y5 } Five point-pairs describing the function

and T5 = Period of the function

OUTPUT (Example



In this example T1=0; if T1≠0, the output is set to Y1 for 0 ≤TIME ≤T1

APPLICATION

This model is a function and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
PULSE GENERATOR	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

PULSE

DESCRIPTION

This model produces a periodic output of pulses of the shape described below.

USAGE

N1 < PULSE(RATE, TD, TR, TL, TF) > N2

where: RATE = frequency of output

TD = delay time

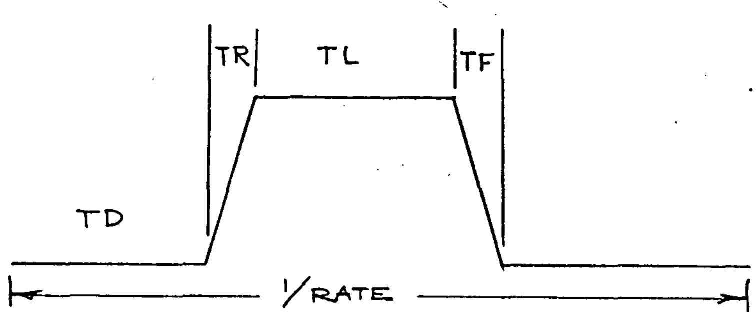
TR = rise time

TL = level time

TF = fall time

OUTPUT

The maximum value is plus one and the minimum value is zero.



APPLICATION

This model is a function and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
SQUARE WAVE GENERATOR	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

SQ
SQUARE WAVE

USAGE

N1 < SQ(RATE) > N2

where: RATE = frequency of the square wave output at node N2

OUTPUT

A square wave of period 1/RATE whose maximum value is plus one and whose minimum value is minus one.

MODEL	GROUP ID	PAGE	DATE
TABLE	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

TABLE

DESCRIPTION

This function provides a piece-wise linear function for modeling both driving functions and non-linearities.

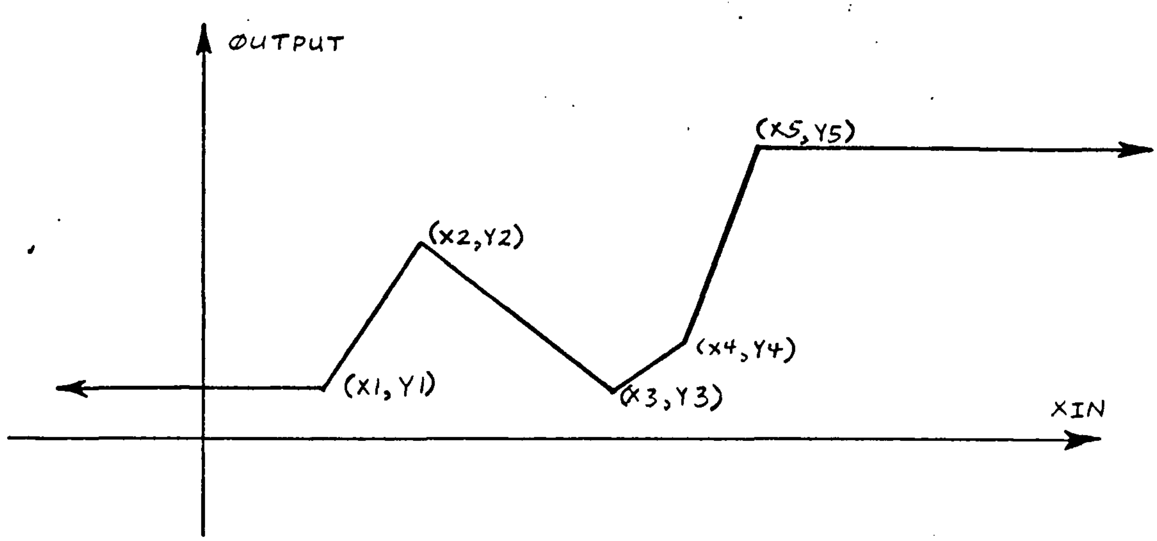
USAGE

N1 < TABLE (XIN, X1, Y1, X2, Y2, X3, Y3, X4, Y4, X5, Y5 > N2

where: XIN = independent variable

X1, Y1
⋮
X5, Y5 } Five point pairs describing the function (Note: Out-of-range values assume the end point values)

OUTPUT (Example)



MODEL	GROUP ID	PAGE	DATE
TABLE	Sig. Gen.	2	April, 1972

APPLICATION

This model is a function and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
TABL2	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

TABL2

DESCRIPTION

This function provides a piece-wise linear function for modeling both driving functions and non-linearities, and provides zero output levels for out of range conditions.

USAGE

N1 <TABL2 (XIN, X1, Y1, X2, Y2, X3, Y3, X4, Y4, X5, Y5)> N2

where: XIN = independent variable

X1, Y1 ⋮ X5, Y5 } five point pairs describing the function

OUTPUT

This model is the same as model "TABLE" except out-of-range values (XIN< X1 or XIN ≥ X5) yield an output of zero.

APPLICATION

This model is a function and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
TRANSCENDENTAL FUNCTIONS	Sig. Gen.	1	April, 1972

LIBRARY MODEL NAMES

SIN	SINE
COS	COSINE
TAN	TANGNT

DESCRIPTION

These elements are FORTRAN transcendental functions or utilize FORTRAN functions.

USAGE (Example)

N1 < SIN ($) > N2

N2 is set to the trigonometric sine of N1

$$\left.\begin{array}{l}\text{SIN }(x)\\ \text{COS }(x)\\ \text{TAN }(x)\end{array}\right\}\ x\text{ in radians}\qquad \left.\begin{array}{l}\text{SINE }(y)\\ \text{COSINE }(y)\\ \text{TANGNT }(y)\end{array}\right\}\ y\text{ in cycles}$$

APPLICATION

These elements are functions and may be used in expressions.

MODEL	GROUP ID	PAGE	DATE
AMPLITUDE MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

AM MODULATOR
AMMOD

DESCRIPTION

The Linear Amplitude Modulator provides classical modulation capability to the ASYSTD user. This element is the baseband model.

USAGE

N1 <AMMOD(BETA,FC)> N2

where: BETA = Modulation Index (ratio)

FC = Carrier frequency

OUTPUT

Let INPUT(t) be the real input to the model (value at node N1) and Vo(t) be the real output of the model, then

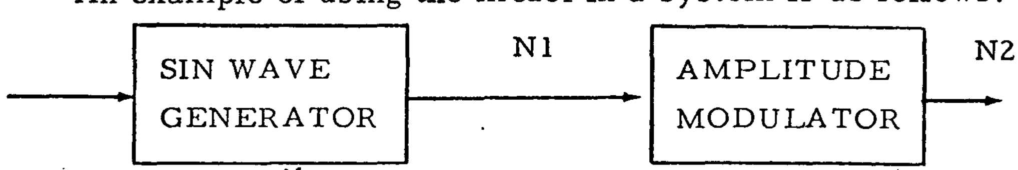
$$Vo(t) = (1.0 + BETA \cdot INPUT(t)) \cdot \cos(2\pi FC \cdot Time)$$

RESTRICTIONS

BETA*INPUT(t) ≤ 1.0 for no over-modulation.

APPLICATION

An example of using the model in a system is as follows:



INPUT < SINE (T) > N1

N1 <AMMOD (1.0, 100 E3)> N2

MODEL	GROUP ID	PAGE	DATE
RF AMPLITUDE MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

RF AMPLITUDE MODULATOR
RAMMOD

DESCRIPTION

The Linear Amplitude Modulator provides classical modulation capability to the ASYSTD user. The output of this model is a complex baseband signal which represents the modulated carrier in the baseband.

USAGE

N1 < RAMMOD (BETA, PHI, MODE) > N2

where: BETA = Modulation Index (ratio)

PHI = Instantaneous Phase Jitter

MODE = 1.0 for Double Sideband (DSB)

MODE = 0.0 for Double Sideband-Suppressed Carrier

OUTPUT

Let i(t) be the real input to the model (node N1),

then: $Vo(t) = \left[MODE + BETA*i_{(t)}\right] e^{j(\omega_c t + PHI)}$

translating Vo(t) to baseband:

$$Vo'(t) = Vo(t)e^{-j\omega_c t}$$

or $Vo'(t) = \left[MODE + BETA*i(t)\right] e^{jPHI} = Vr(t) + jVi(t)$

where: $Vo'(t)$ is the complex baseband output signal at N2

with $Vr(t) = \left[MODE + BETA*i(t)\right] COS(PHI)$

$Vj(t) = \left[MODE + BETA*i(t)\right] SIN(PHI)$

MODEL	GROUP ID	PAGE	DATE
RF AMPLITUDE MODULATOR	Modulator	2	April, 1972

RESTRICTIONS

$|BETA*i(t)| \leq 1.0$ for no over-modulation

APPLICATION

This model is used in place of the amplitude modulator when carrier translation is required.

MODEL	GROUP ID	PAGE	DATE
LINEAR FREQUENCY MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

FM MODULATOR
FMMOD

DESCRIPTION

The Linear Frequency Modulator Model provides a classical model for this type of angle modulation. The carrier output magnitude is defined as unity.

USAGE

N1 < FMMOD(DF, FC) > N2

where: DF = frequency deviation (Hz) of the carrier per unit input

FC = carrier frequency

OUTPUT

Let the signal at N1 at time T be i(t).
Let the signal at N2 at time T be Vo(t).

then: $$Vo(t) = SINE(FC*t + DF*\int_{TSTART}^{t} i(\tau)\, d\tau)$$

NOTE: The arguments of the SINE function is in Hz.

APPLICATION

FMMOD is for use when no carrier translation is required in the simulation.

MODEL	GROUP ID	PAGE	DATE
RF LINEAR FREQUENCY MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

RF FM MODULATOR
RFMMOD

DESCRIPTION

The Linear Frequency Modulator model provides a classical model for this type of angle modulation. The carrier output magnitude is defined as unity.

USAGE

N1 < RFMMOD(DF) > N2

where: DF = frequency deviation (Hz) of the carrier per unit input.

OUTPUT

Let i(t) be the real input to the model (node N1), then for the ideal FM modulator

$$Vo(t) = e^{j(\omega_c t + \beta\int_o^t i(t)dt)}$$

where: $\beta = Z \pi DF$

translating Vo(t) to baseband:

$$Vo'(t) = Vo(t)\, e^{-j\omega_c t}$$

or

$$Vo'(t) = e^{j\beta\int_o^t i(t)dt} = Vr(t) + jVj(t)$$

where: $Vo'(t)$ is the complex baseband output signal at N2

with: $Vr(t) = COS\ (\beta\int_o^t i(t)dt)$

$$Vj(t) = SIN\ (\beta\int_o^t i(t)dt)$$

APPLICATION

This model is used in place of the FREQUENCY MODULATOR model when carrier translation is required in the simuation.

MODEL	GROUP ID	PAGE	DATE
SQUARE WAVE FREQUENCY MODULATOR	Modulators	1	

LIBRARY MODEL NAMES

SQUARES WAVE FREQUENCY MODULATOR
SQFMOD

DESCRIPTION

The Square Wave Frequency Modulator provides a model for this type of angle modulation with ideal limiting.

USAGE

N1 < SQFMOD(DF, FC) > N2

where: DF = frequency deviation (cycles) of the carrier per unit input

FC = carrier frequency

OUTPUT

Let the signal at N1 at time T be INP(T).
Let the signal at N2 at time T be OUT(T).

Then: $$F(T) = \text{SINE}(FC*T + DF*\int_{TSTART}^{T} \text{INP}(t)dt) \quad \begin{bmatrix}\text{linear frequency} \\ \text{modulation}\end{bmatrix}$$

NOTE: The arguments of the SINE function is in cycles

OUT(T) = +1 when F(T) ≥ 0
OUT(T) = -1 when F(T) < 0

MODEL	GROUP ID	PAGE	DATE
LINEAR PHASE MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

PHASE MODULATOR
PMMODD
PMMOD

DESCRIPTION

The Linear Phase Modulator provides a classical model for this type of angle modulation. The carrier output level is defined as unity.

USAGE

N1 <PMMOD(BETA, TC)> N2

where: BETA = Phase (Radians) deviation per unit input

FC = Carrier frequency (Hz)

NOTE: Modulation Index = BETA * Input_{max}

OUTPUT

Let the signal at N1 at time T be i(t)
Let the signal at N2 at time T be Vo(t)

then: $Vo(t) = \sin(2\pi * FC * t + BETA * i(t))$

APPLICATION

This model is for use when no carrier translation is required in the simulation.

MODEL	GROUP ID	PAGE	DATE
RF LINEAR PHASE MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

RF PHASE MODULATOR
RPMMOD

DESCRIPTION

The Liner Phase Modulator provides a classical model for this type of angle modulation. The output is the complex baseband signal.

USAGE

N1 < RPMMOD(BETA) > N2

where: BETA = Phase (Radians) deviation per unit input

OUTPUT

Let i(t) be the real input to the model, then:

$$Vo(t) = e^{j\left\{\omega_c t + \beta \cdot i(t)\right\}}$$

translating Vo(t) to baseband:

$$Vo'(t) = Vo(t)e^{-j\omega_c t}$$

or

$$Vo'(t) = e^{j\beta \cdot i(t)} = V_r(t) + jV_j(t)$$

where: $Vo'(t)$ is the complex baseband output signal at N2

with $Vr(t) = COS(\beta \cdot i(t))$

$Vj(t) = SIN(\beta \cdot i(t))$

APPLICATION

This model is used in place of the FM MODULATOR model when carrier translation is required in the simulation.

MODEL	GROUP ID	PAGE	DATE
SQUARE WAVE PHASE MODULATOR	Modulator	1	April, 1972

LIBRARY MODEL NAMES

SQUARE WAVE PHASE MODULATOR
SQPMOD

DESCRIPTION

The Square Wave Phase Modulator provides a model for this type of angle modulation with ideal limiting.

USAGE

N1 < SQPMOD(BETA, FC) > N2

where: BETA = Phase (Radians) deviation per unit input

FC = Carrier frequency

NOTE: Modulation Index = $BETA*Input_{max}$

OUTPUT

Let the signal at N1 at time T be INP(T).
Let the signal at N2 at time T be OUT(T).

$F(T) = \sin(2\pi*FC*T + BETA*INP(T))$ [linear phase modulation]

$OUT(T) = +1$ when $F(T) \geq 0$

$OUT(T) = -1$ when $F(T) < 0$

MODEL	GROUP ID	PAGE	DATE
DELTA MODULATION	Modulator	1	April, 1972

LIBRARY MODEL NAMES

DELTA MODULATOR
DELMOD

DESCRIPTION

Delta modulation is a coded modulation system nearly as efficient as PCM, requires more bandwidth than PCM, but has much simpler circuitry. These advantages make delta modulation quite attractive as a standard model. The output waveform magnitude is defined as ± 1.

USAGE

N1 <DELMOD(PW,PPS)> N2

where: PW = Pulse width (unit time)

PPS = Pulse repetition rate (pulses/unit time)

OUTPUT

In a delta modulation system, only the changes in signal amplitude from sample to sample are output. The process consists of utilizing a pulse generator (clock), one shot multi-vibrator, an integrator, and a difference circuit.

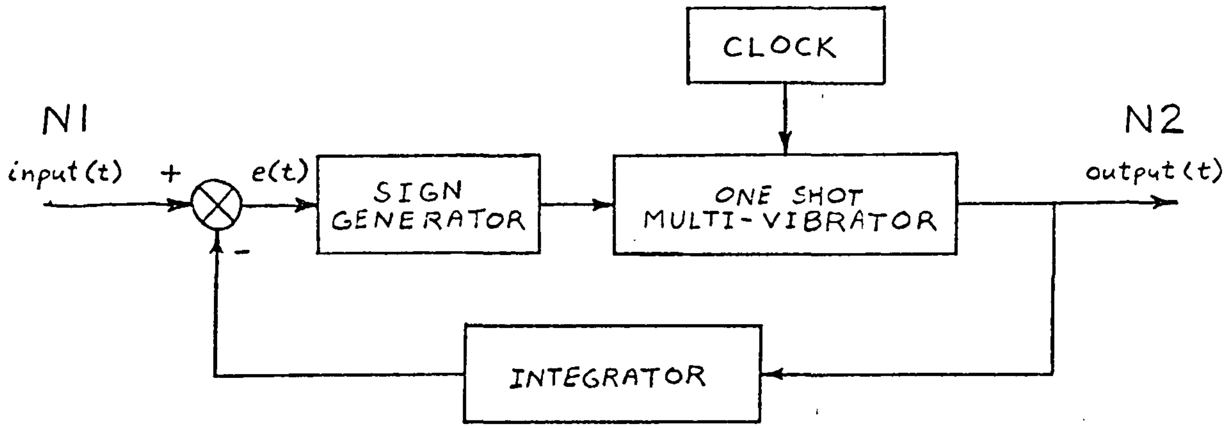
$$\text{let } e_{(t)} = \text{input}_{(t)} - \int \text{Output}_{(t)}\, dt$$

$$\text{where output}_{(t)} = \text{Sign}\,(e_{(t)}) * \delta(t)$$

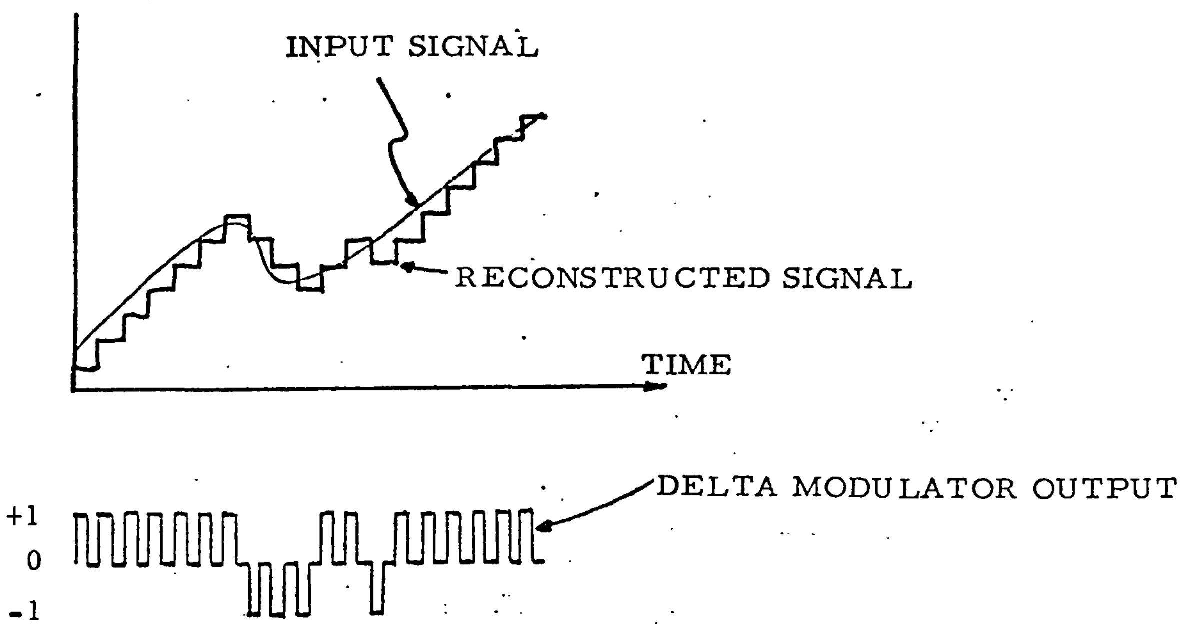
where δ (t) is a finite pulse of width PW

The above process is clocked at a repetition rate of PPS

MODEL	GROUP ID	PAGE	DATE
DELTA MODULATION	Modulator	2	April, 1972



EXAMPLE:



The width of the pulses in the Delta Modulator output is PW. There is one positive or negative pulse every 1/PPS. The input signal and the signal which can be reconstructed from the Delta Modulator output have the same scale. Each step is 1/PPS wide and PW high.

MODEL	GROUP ID	PAGE	DATE
DELTA MODULATION	Modulator	3	April, 1972

RESTRICTIONS

In general, the delta modulator cannot follow the input signal whenever:

$$\left| \frac{d}{dt} \text{ input}(t) \right| > PW*PPS$$

To prevent overload, PW and PPS should be adjusted so that:

$$\left| \frac{d}{dt} \text{ input}(t) \right|_{max} \leq PW*PPS$$

which is usually satisfied if:

$$A\omega \leq PW*PPS$$

where: A = Peak value of input

ω = Maximum frequency of input

If PW and PPS cannot be adjusted to meet this condition (note $PWS \leqslant 1/PPS$), then the input to the delta modulator must be properly scaled.

In the case where:

$$\left| \frac{d}{dt} \text{ input}(t) \right|_{max} << PW*PPS$$

PW may be decreased or the input signal scaled to provide more information in the delta modulator output.

APPLICATION

The delta modulator is normally used in pulse coding an audio signal for subsequent transmission via a modulated carrier. As such, this model will be mainly used in generating a baseband signal.

REFERENCE

For a description of delta modulation and a bibliography, see:

H. R. SCHINDLER, "Delta Modulation", IEEE SPECTRUM, October 1970, pp. 69-78

MODEL	GROUP ID	PAGE	DATE
AMPLITUDE DEMODULATOR	Demod.	1	April, 1972

LIBRARY MODEL NAMES

AMPLITUDE DEMODULATOR
AMDEM

DESCRIPTION

This linear Amplitude Demodulator provides a rudimentary model for use in the ASYSTD library. The basic model is a full wave rectifier followed by a user selected filter function chosen for the particular application.

USAGE

N1 < AMDEM > N2

NOTE: N2 must be the input to a filter.

OUTPUT

The output signal at N2 is simply the absolute value of the signal at N1

$$N2(T) = |N1(T)|$$

APPLICATION

This model is for use in the baseband region. Its output must be fed through an averaging filter to eliminate the carrier.

MODEL	GROUP ID	PAGE	DATE
RF AMPLITUDE DEMODULATOR	Demod.	1	April, 1972

LIBRARY MODEL NAMES

RF AM DEMODULATOR
RAMDEM

DESCRIPTION

This model is a rudimentary Amplitude Demodulator for use in modeling in the RF region.

USAGE

N1 < RAMDEM(GAIN) > N2

where: GAIN = Amplification of the model (typically 1/BETA, see RF AMPLITUDE MODULATOR)

OUTPUT

Let X be the complex signal at N1

$X = X_r + jXi$

$|X| = \left(X_r^2 + X_i^2\right)^{1/2}$

Let Y be the output at N2

$Y = GAIN* \left(|X| - 1.\right)$

APPLICATION

This model should be followed with a user selected filter for accurate simulation.

MODEL	GROUP ID	PAGE	DATE
FREQUENCY DEMODULATOR WITH FEEDBACK	Demod.	1	April, 1972

LIBRARY MODEL NAMES

FMFB

DESCRIPTION

The Frequency Demodulator with Feedback Model (FMFB) provides an alternate demodulation process capability. The basic model consists of a multiplier, IF filter, FM discriminator (FMDEMOD) and a Voltage Controlled Oscillator (FMMOD). The RF filter and post detection low pass filter are external to the model.

USAGE

N1 <(FMFB (NIF,NTYPE,AR,EM,BIF,FIF,GAIN,FC,DV,DF)> N2

where: NIF -IF Filter Order (≤ 10)

NTYPE - Type of Filter Function:
- = 1 for Butterworth
- = 2 for Chebyshev
- = 3 for Bessel
- = 4 for Butterworth-Thomson
- = 5 for Elliptic

AR -Amplitude Ripple (dB)

EM -M-Factor for Butterworth-Thomson
Stop Band Ratio for Elliptic (if positive)
Modular Angle (Degrees) for Elliptic (if negative)

BIF -IF Filter Bandwidth

GAIN -Detector Gain + VCO Amp Gain

FIF -IF Frequency

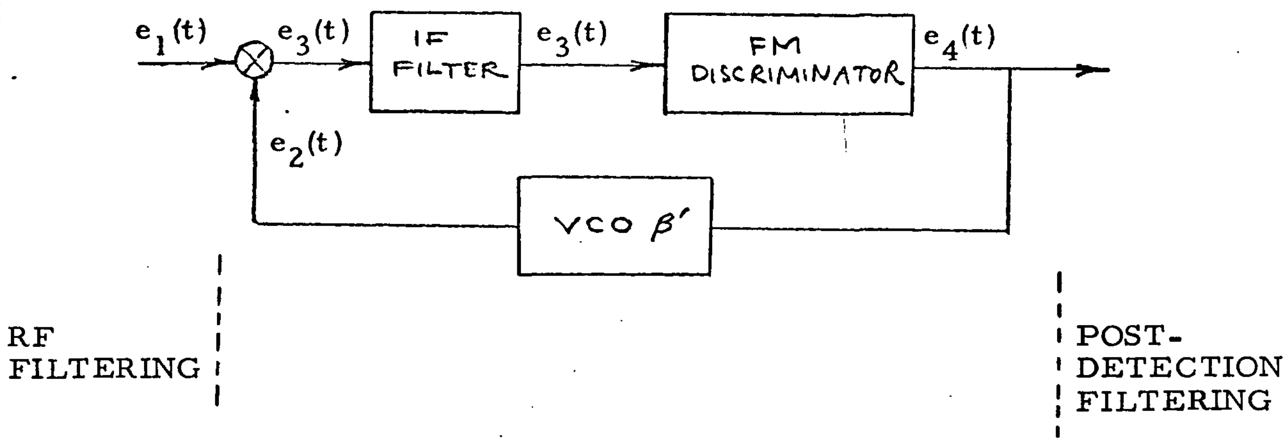
FC -Carrier Frequency

DV -FM Discriminator Constant (Volts/Hz)

DF -VCO Deviation (e.g., Hz/Volts)

NOTE: N1 is the output of an RF filter and N2 is the input to a low pass filter.

DETAILED DESCRIPTION



Let

$$e_1(t) = A(t)\cos(\omega_{c} t + \phi_i(t))$$

and

$$e_2(t) = -B\sin(\omega_{VCO} t + \theta(t))$$

where

$$\phi_i(t) = \beta \int S(t)$$

$$\theta(t) = \beta' \int e_4(t)$$

and

$$e_3'(t) = \frac{A(t)B}{2}\Big\{ \sin[\omega_{IF} t + \phi_i(t) - \theta(t)] - \sin[(\omega_{IF} + \omega_{c})t + \phi(t) + \theta(t)]\Big\}$$

assuming the IF filter passes only the first term

$$e_3'(t) = \frac{A(t)B}{2} \sin (\omega_{IF}t + \phi_i(t) - \theta(t))$$

the output of the FM discriminator is ideally

$$e_4(t) = \frac{d}{dt} [\phi_i(t) - \theta(t)] DV = DV [\beta S(t) - \beta' e_4(t)]$$

$$e_4(t) = DV \frac{\beta}{1 + \beta'} S(t)$$

APPLICATION

The FMFB demodulator utilizes a tracking principle to achieve good SNR performance.

MODEL	GROUP ID	PAGE	DATE
PHASE DEMODULATOR	Demod.	1	April, 1972

LIBRARY MODEL NAMES

PHASE DEMODULATOR
PMDEMM

DESCRIPTION

This Phase Demodulator simply takes the integral of an FM demodulator output.

USAGE

N1 < PMDEMM(DV, FC) > N2

where: FC = Center Frequency

DV = Output Magnitude per Unit Phase Deviation (Volts/Radians)

OUTPUT

The Phase Demodulator output is given by $DV * \int FMDEMOD(t)\, dt$ where FMDEMOD(t) is the output of an FMDEMOD with a sensitivity of 1v/radian.

APPLICATION

This model is to be used with external filtering.

MODEL	GROUP ID	PAGE	DATE
RF PHASE DEMODULATOR	Demod.	1	April, 1972

LIBRARY MODEL NAMES

RF PHASE DEMODULATOR
RFPDEM

DESCRIPTION

This model represents an ideal wide band phase demodulator.

USAGE

N1 <RFPDEM(DV)> N2

where: DV = Output Magnitude per Unit Phase Deviation (Volts/Radians)

OUTPUT

Let the complex input at N1 be $X = X_r + jX_i$

Then the output of this model is $DV*TAN^{-1}\left(\frac{Xi}{X_r}\right)$

APPLICATION

This model is to be used with external filtering.

SYSTEMS ASSOCIATES

MODEL	GROUP ID	PAGE	DATE
FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

FILTER

DESCRIPTION

The modeling of filters, or continuous functions relies on several computer routines previously developed by SAI which perform the various functions described in Appendices A and C. For ease in their use, an interface routine is written called FILTER, with several entry points as is explained below.

When utilizing any Filter model in a simulation of an RF link, translation of the filter to the baseband region is necessary for efficient simulation. The translation parameters are reflected in the reference to the Filter model.

DETAILED DESCRIPTION

The detailed description for generating the various filter functions in the s domain is described in Appendix A. Once the function of s is known, the bilinear z transform is derived. In order to reduce round-off errors, the function is represented by second degree sections, or quadratic factors. The bilinear z-transform converts a factor of s to a factor of the same degree in z, that is:

$$\frac{O(s)}{I(s)} = \frac{a_2 s^2 + a_1 s + a_o}{b_2 s^2 + b_2 s + b_o} \Bigg| \frac{F_2 z^{-2} + F_1 z^{-1} + F_o}{D_2 z^{-2} + D_1 z^{-1} + D_o} = \frac{O(z)}{I(z)}$$

NOTE: D_o is normalized to entry

z^{-1} is a unit delay

The difference equation for one of the quadratic factors will then be:

$$O(t) = F_2 I(t - 2DT) + F_1 I(t - DT) + F_o I(t) - D_2 O(t - 2DT) - D_1 O(t - DT)$$

where: DT is the sampling time.

If the filter is not being translated, the quadratic factors are cascaded. However, when translating the filter (described in Appendix A), both real and imaginary coefficients of s result and the function is represented by parallel quadratic factors. The representation of the function as a sum of terms rather than a product eliminates the necessity of computing the roots of a polynominal in determining $K_r(s)$ and $K_i(s)$, (see Appendix A).

When using a translated filter, the run time can be reduced significantly in trade for exact representation of the filter. Reduction of approximately one-fourth is realized by using an equivalent low pass function; or one-half by assuming symmetry of the filter (i.e., $K_i(s) = 0$). This is accomplished when referencing one of the functions as described below.

USAGE

N1 <FILTER(NP,IF,IG,FX,BW,FC,AMP,AR,EM)> N2

All variables must be included whether they are applicable or not.

where:
- NP = filter order
- IG = filter geometry
  - = 1 for Low Pass
  - = 2 for High Pass
  - = 3 for Band Pass
  - = 4 for Band Stop
- AR = amplitude ripple (dB)
- EM = M-factor for Butterworth-Thomson or stop-band ratio (>0) or modulator angle (<0) for Elliptic functions
- FX = arithmetic center frequency
- BW = bandwidth
- FC = translation frequency (i.e., translate such that FC becomes zero)
- AMP = voltage gain at FX

IF = filter function
= 1 for Butterworth
= 2 for Chebyshev
= 3 for Bessel
= 4 for Butterworth-Thomson
= 5 for Elliptic

Alternate references to filters are as follows:

BUTTERWORTH (NP, IG, FX, BW, FC, AMP)

CHEBYSHEV (NP, IG, FX, BW, FC, AMP, AR)

BESSEL (NP, IG, FX, BW, FC, AMP)

BUTTERWORTH THOMSON (NP, IG, FX, BW, FC, - AMP, EM)

ELLIPTIC (NP, IG, FX, BW, FC, AMP, AR, EM)

Special Cases:

a) A model is available to characterize a filter from frequency response data (see GENERAL FILTER).

b) QFACTOR (AMP, A1, A2, A3, A4, A5, A6)

used to describe:

$$AMP * \frac{A1s^2 + A2s + A3}{A4s^2 + A5s + A6}$$

c) LEADLAG (AMP, F1, F2, F3, F4)

used to describe:

$$AMP * \frac{\left(\frac{s}{2\ F1} + 1\right)\left(\frac{s}{2\ F2} + 1\right)}{\left(\frac{s}{2\ F3} + 1\right)\left(\frac{s}{2\ F4} + 1\right)}$$

if:

F2 = 0 then one zero is eliminated
F1 = 0 then both zeros are eliminated
F4 = 0 then one pole is eliminated
F3 = 0 then both poles are eliminated

d) LEAD FUNCTION (AMP, F1, F2, F3)

used to describe:

$$AMP * \frac{\left(\frac{s}{2\ F1} + 1\right)\left(\frac{s}{2\ F2} + 1\right)}{\left(s\frac{s}{2\ F3} + 1\right)}$$

and is otherwise the same as the LEADLAG function.

APPLICATIONS AND RESTRICTIONS

When utilizing the above function, any time FC>0, an RF filter is referenced (i.e., complex inputs and outputs) rather than a baseband filter (i.e., real inputs and outputs). The following table describes the conditions set up by FC and FX.

FC	FX	IG	Result
0	-	--	Baseband filter simulation
>0	>0	3	RF translated filter
>0	<0	3	Symmetric translated filter (Q = ∞)
>0	0	≠ 1	Equivalent low pass function

MODEL	GROUP ID	PAGE	DATE
BUTTERWORTH FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

BUTTERWORTH
BUTWTH
BUFUNCTION

DESCRIPTION

For a description of Butterworth filters, see Appendix C, Section C.2.2.

USAGE

N1 <BUTTERWORTH(NP, IG, FX, BW, FC, AMP)> N2

NP = filter order
IG = filter geometry
= 1 for Low Pass
= 2 for High Pass
= 3 for Band Pass
= 4 for Band Stop
FX = arithmetic center frequency
BW = bandwidth
FC = translation frequency (i.e., translate such that FC becomes zero)
AMP = voltage gain at FX

APPLICATION

For applications and restrictions in using this model, see the discussion on the model FILTER.

MODEL	GROUP ID	PAGE	DATE
CHEBYSHEV FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

CHEBYSHEV
CHEBY
TCHEBYCHEFF

DESCRIPTION

For a description of Chebyshev filters, see Appendix C, Section C. 2. 3.

USAGE

N1 <CHEBYSHEV(NP, IG, FX, BW, FC, AMP, AR)> N2

where:
- NP = filter order
- IG = filter geometry
  - = 1 for Low Pass
  - = 2 for High Pass
  - = 3 for Band Pass
  - = 4 for Band Stop
- AR = amplitude ripple (dB)
- FX = arithmetic center frequency
- BW = bandwidth
- FC = translation frequency (i. e., translate such that FC becomes zero)
- AMP = voltage gain at FX

APPLICATION

For applications and restrictions in using this model, see the discussion on the model FILTER.

MODEL	GROUP ID	PAGE	DATE
BESSEL FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

BESSEL
BE FUNCTION

DESCRIPTION

For a description of Bessel filters, see Appendix C, Section C.2.4.

USAGE

N1 <BESSEL(NP, IG, FX, BW, FC, AMP)> N2

NP = filter order

IG = filter geometry
 = 1 for Low Pass
 = 2 for High Pass
 = 3 for Band Pass
 = 4 for Band Stop

FX = arithmetic center frequency

BW = bandwidth

FC = translation frequency (i.e., translate such that FC becomes zero)

AMP = voltage gain at FX

APPLICATION

For applications and restrictions in using this model, see the discussion on the model FILTER.

MODEL	GROUP ID	PAGE	DATE
BUTTERWORTH THOMSON FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

BUTTERWORTH THOMSON
BT FUNCTION
BUTOM

DESCRIPTION

For a description of Butterworth Thomson filters, see Appendix C, Section C.2.5.

USAGE

N1 <BUTTERWORTH THOMSON(NP, IG, FX, BW, FC, AMP, EM)> N2

NP = filter order
IG = filter geometry
= 1 for Low Pass
= 2 for High Pass
= 3 for Band Pass
= 4 for Band Stop
FX = arithmetic center frequency
BW = bandwidth
FC = translation frequency (i.e., translate such that FC becomes zero)
AMP = voltage gain at FX
EM = M-factor

APPLICATION

For applications and restrictions in using this model, see the discussion on the model FILTER.

MODEL	GROUP ID	PAGE	DATE
ELLIPTIC FUNCTION FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

ELLIPTIC
ELIPTC
EL FUNCTION

DESCRIPTION

For a description of Elliptic Function Filters, see Appendix C, Section C.2.6.

USAGE

N1 < ELLIPTIC(NP, IG, FX, BW, FC, AMP, AR, EM)> N2

where:
- NP = filter order
- IG = filter geometry
  - = 1 for Low Pass
  - = 2 for High Pass
  - = 3 for Band Pass
  - = 4 for Band Stop
- FX = arithmetic center frequency
- BW = bandwidth
- FC = translation frequency (i.e., translate such that FC becomes zero)
- AMP = voltage gain at FX
- AR = amplitude ripple (dB)
- EM = stop-band ratio if positive modular angle if negative

APPLICATION

For applications and restrictions in using this model, see the discussion on the model FILTER.

MODEL	GROUP ID	PAGE	DATE
GENERAL FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

GENERAL FILTER
GENRAL

DESCRIPTION

The General Filter model determines an arbitrary element transfer function by the complex curve fitting method. The procedure is entirely analogous to familiar curve fitting techniques except that the quantities involved are complex numbers. It requires an assumption on the part of the user as to the number of poles and zeros which characterize the transfer function of the element in question. From an appropriate number of representative amplitude and phase response samples over the frequency range of interest, the values of these complex poles and zeros which characterize the element transfer function may then be determined.

DETAILED DESCRIPTION

(See also Appendix C)

A generalized element transfer function H(s) is customarily written in the form:

$$H(s) = \frac{A \prod_{i=1}^{NZ} (s - Z_i)}{\prod_{i=1}^{NP} (s - P_i)} \qquad (1)$$

where:
- s = $j\omega$, complex frequency
- P_i = complex pole $(s + j\omega_i)$
- Z_i = complex zero $(s + j\omega_i)$
- NP = number of poles (also filter order)
- NZ = number of zeros
- A = multiplicative real constant

The poles and zeros are always either complex conjugate pairs or single real values.

For each response sample at frequency ω_k the poles and zeros are related to the empirically determined amplitude and phase through

$$\frac{A \prod_{i=1}^{NZ} (j\omega_k - Z_i)}{\prod_{i=1}^{NP} (j\omega_k - P_i)} = \alpha_k + j\beta_k \qquad (2)$$

where

$$\text{Magnitude response} = \sqrt{\alpha^2 + \beta^2}$$

$$\text{Phase response} = \tan^{-1} \beta/\alpha$$

Hence, for each response sample point, the right side of Equation (2) is determined and the left side is a complex expression in the P_i's and Z_i's. If the form of H(s) is assumed by specifying the number of poles and zeros, the P_i's, Z_i's, and A are determined by the set of NP+NZ+1 such complex equations which result from the substitution of corresponding phase and amplitude response samples at a like number of frequencies. The actual solution of this system of complex linear equations is effected by standard library subroutines.

USAGE

N1 <GENRAL(NP, NZ, IDB, FC, IG, BW, AMP)> N2

where:
- NP = assumed number of poles
- NZ = assumed number of zeros
- IDB = 1 if response amplitude data is to be in dB
  - = 0 if response amplitude data is normalized to 1.0
- FC = translation frequency (i.e., translate such that FC becomes zero)

IG = filter geometry
= 1 for Low Pass
= 2 for High Pass
= 3 for Band Pass
= 4 for Band Stop

BW = bandwidth

AMP = voltage gain (ratio) mid-way between the upper and lower cutoff frequencies

INPUT

The number of frequency response data points should equal NP+NZ+1.

Input frequency response data is read in through the subroutine POLZER, one card per response data point, in the order: Frequency (Hz), Amplitude (dB or relative magnitude), Phase (degrees). The input data format is (1PE15.4, OP2F15.4). An input data set for a general 3rd order filter with two zeros (NP+NZ+1 = 6) is as follows:

FREQUENCY	AMPLITUDE	PHASE
1.4320-05	.0000	-10.3210
4.7750-05	-.0030	-34.9240
1.0980-04	-.4420	-87.3390
1.6710-04	-3.6820	-141.9130
2.8970-04	-15.7100	157.0050
3.3740-04	-19.6160	146.5060

MODEL	GROUP ID	PAGE	DATE
QUADRATIC FACTOR	Filter	1	April, 1972

LIBRARY MODEL NAMES

QFACT
QFACTOR
QUADRADIC FACTOR

DESCRIPTION

This model is used to describe the transfer function:

$$H(s) = AMP * \frac{A1s^2 + A2s + A3}{A4s^2 + A5s + A6}$$

(see Appendix C, Section C. 1)

USAGE

N1 <QFACTOR(AMP, A1, A2, A3, A4, A5, A6)> N2

where: AMP = amplification constant

A1-A6 = constants specifying the transfer function

MODEL	GROUP ID	PAGE	DATE
LEAD LAG	Filter	1	April, 1972

LIBRARY MODEL NAMES

LEAD LAG
LOOP FILTER

DESCRIPTION

This model is used to describe the transfer function:

$$H(s) = AMP * \frac{\left(\frac{s}{2\pi F1}+1\right)\left(\frac{s}{2\pi F2}+1\right)}{\left(\frac{s}{2\pi F3}+1\right)\left(\frac{s}{2\pi F4}+1\right)}$$

(see Appendix C, Section C.1)

USAGE

N1 <LEADLAG(AMP, F1, F2, F3, F4)> N2

where: AMP = amplification constant
F1-F4 = constants specifying the transfer function, and if:

F2 = 0 then one zero is eliminated
F1 = 0 then both zeros are eliminated
F4 = 0 then one pole is eliminated
F3 = 0 then both poles are eliminated

MODEL	GROUP ID	PAGE	DATE
LEAD FUNCTION	Filter	1	April, 1972

LIBRARY MODEL NAMES

LEAD FUNCTION

DESCRIPTION

This model is used to describe the transfer function

$$H(s) = AMP * \frac{\left(\frac{s}{2\ F1} + 1\right)\left(\frac{s}{2\ F2} + 1\right)}{\left(s\frac{s}{2\ F3} + 1\right)}$$

(see Appendix C, Section C. 1)

USAGE

N1 <LEAD FUNCTION(AMP, F1, F2, F3)> N2

where: AMP = amplification constant

F1-F3 = constants specifying the transfer function, and if:

F2 = 0 then one zero is eliminated

F1 = 0 then both zeros are eliminated

F3 = 0 the pole is eliminated

MODEL	GROUP ID	PAGE	DATE
MATCHED FILTER	Filter	1	April, 1972

LIBRARY MODEL NAMES

MATCHED FILTER
MFLTER

DESCRIPTION

The Matched Filter model is a simple integrate and dump routine clocked to the Bit Time.

USAGE

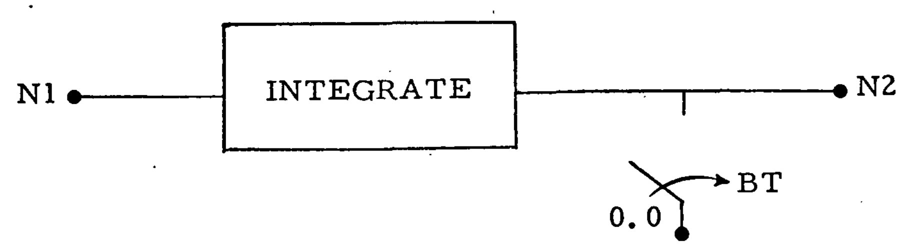
N1 <MFLTER(BT)> N2

where: BT = bit time

OUTPUT

The output at N2 is the trapezoidal approximation of the integral of the signal at N1. Every BT, this integral is reset to zero.

BLOCK DIAGRAM



RESTRICTIONS

The integrate-Dump process is asynchronous and starts at time equal to zero, requiring the user's discretion in its use.

MODEL	GROUP ID	PAGE	DATE
SOFT LIMITER	Limiter	1	April, 1972

LIBRARY MODEL NAMES

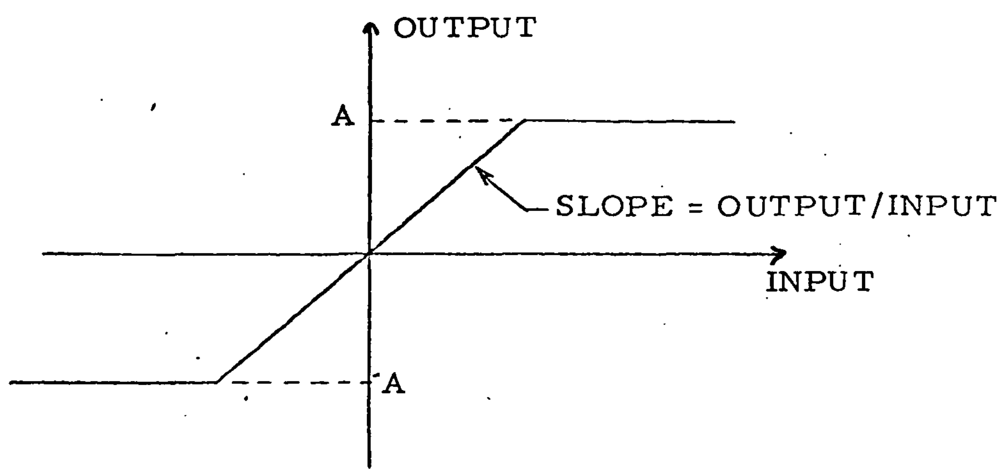
SOFT LIMITER
SOFTY

USAGE

N1 <SOFTY(A,SLOPE)> N2

where: A = maximum amplitude of output
SLOPE = slope of transition limit

OUTPUT



APPLICATION

This element is for use in baseband modeling.

MODEL	GROUP ID	PAGE	DATE
HARD LIMITER	Limiter	1	April, 1972

LIBRARY MODEL NAMES

HARD LIMITER
HARD

USAGE

N1 <HARD> N2

OUTPUT

The absolute value of the output at N2 is 1. The sign of the output is the same as the input at N1.

APPLICATION

This element is for use in baseband modeling.

MODEL	GROUP ID	PAGE	DATE
RF SOFT LIMITER	Limiter	1	April, 1972

LIBRARY MODEL NAMES

RF SOFT LIMITER
RFSOFT

USAGE

N1 <RFSOFT(A,SLOPE)> N2

where: A = maximum amplitude of output
SLOPE = slope of transition limit

OUTPUT

Let $X = X_r + jX_i$ be the complex signal at N1
and Let $Y = Y_r + jX_i$ be the complex signal at N2

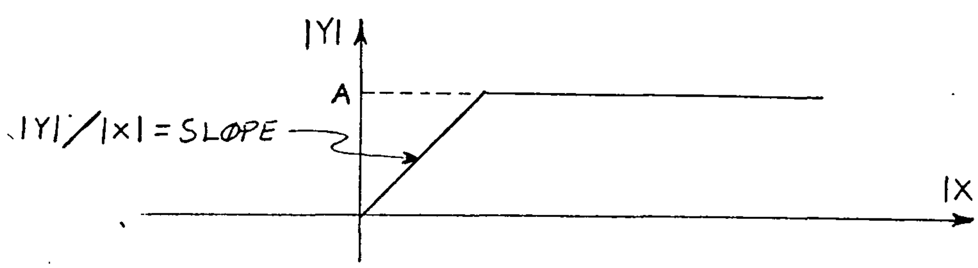
$$|X| = (X_r^2 + X_i^2)^{1/2} = \text{magnitude of } X$$

$$\varnothing(X) = \tan^{-1}(X_i/X_r) = \text{phase of } X$$

Then the output at N2 is such that:

1) $\varnothing(Y) = \varnothing(X)$

2) $|Y| = F(|X|)$ as described by the graph below:



APPLICATION

This element is for modeling in the RF region.

MODEL	GROUP ID	PAGE	DATE
RF HARD LIMITER	Limiter	1	April, 1972

LIBRARY MODEL NAMES

RF LIMITER
RFLIMT

USAGE

N1 <RF LIMITER> N2

OUTPUT

Assume the input at N1 is described by:

$$X = X_r + jX_i = \left[X_r^2 + X_i^2\right]^{1/2} e^{j \tan^{-1} (X_i/X_r)}$$

then the output at N2 is:

$$Y = Y_r + jY_i = C e^{j \tan^{-1} (Y_i/Y_r)}$$

where:

$$Y_r = \frac{CX_r}{\left[X_r^2 + X_i^2\right]^{1/2}}$$

$$Y_i = \frac{CX_i}{\left[X_r^2 + X_i^2\right]^{1/2}}$$

In this model C = 1.

In other words, the phase of the signal is maintained but the magnitude is set to 1.

APPLICATION

The element is for use in modeling in the RF region.

MODEL	GROUP ID	PAGE	DATE
FOURIER TRANSFORM	Transforms	1	April, 1972

LIBRARY MODEL NAMES

FOURT	IFOURT
FOURIER	INVERSE FOURIER

DESCRIPTION

This model samples an input signal and calculates the Discrete Fourier Transform of these samples. Its output is the coefficients of this transform. IFOURT performs the inverse transform, reconstructing the input signal to FOURT.

USAGE

N1 <FOURT(N, TW)> N2

or

N1 <IFOURT(N, TW)> N2

where: N = number of samples per transform

TW = time window per transform

NOTE: Although N can be any positive integer, the transform is much faster if all the prime factors of N are 2, 3 or 5.

NOTE: The actual time window used is the multiple of N*DT nearest TW.

OUTPUT

Each TW, this model takes N samples of the input at N1, and calculates the Discrete Fourier Transform (or the inverse of the transform). The output of the model during this TW is the N transformed values produced during the previous TW.

NOTE: The input and the output of this model are complex.

MODEL	GROUP ID	PAGE	DATE
FOURIER TRANSFORM	Transforms	2	April, 1972

REFERENCE

For a discussion of this transform, see COMSAT Laboratories, "Orthogonal Transform Feasibility Study, Final Report", Contract NAS9-11240, November 1971, Section 3.2.

APPLICATION

The models INTLEV and DELEAV have been provided to deal with the complex inputs and outputs of this model. During the first TW, FOURT outputs a sync pulse of all ones to lock in DELEAV and IFOURT, compensating for any delays in the system.

MODEL	GROUP ID	PAGE	DATE
HAAR TRANSFORM	Transforms	1	April, 1972

LIBRARY MODEL NAMES

HAAR

IHAAR
INVERSE HAAR

DESCRIPTION

This model samples a signal and outputs the HAAR Transform of these samples. (IHAAR-Inverse HAAR Transform - reconstructs the signal given the transformed values.)

USAGE

N1 <HAAR(N, LOG2N, TS)> N2

or N1 <IHAAR(N, LOG2N, TS)> N2 for the inverse

where: N = number of samples per transform (N must be a power of 2)

LOG2N = log base 2 of N, $N = 2^{LOG2N}$

TS = time window per transform

NOTE: The actual time window used is the integer multiple of N*DT nearest TS.

OUTPUT

Each TS, this model takes N samples of its input, and takes the transform of these samples (or the inverse transform), producing N transformed values. The output of this model during this time window is the N transformed values calculated during the previous TS.

REFERENCE

For a discussion of HAAR functions and the HAAR Transform, see COMSAT Laboratories, "Orthogonal Transform Feasibility Study, Final Report", Contract NAS9-11240, November 1971, Section 3.4.

MODEL	GROUP ID	PAGE	DATE
HAAR TRANSFORM	Transforms	2	April, 1972

APPLICATION

The output of the HAAR Transform during the first TS is +1. IHAAR uses this string of ones as a sync pulse to lock on to the transformed coefficients.

MODEL	GROUP ID	PAGE	DATE
HADAMARD TRANSFORM	Transform	1	April, 1972

LIBRARY MODEL NAMES

HDMRD HADAMARD	IHDMRD INVERSE HADAMARD

DESCRIPTION

This model samples a signal and outputs the HADAMARD Transform of these samples. (IHDMRD-Inverse HADAMARD Transform-reconstructs the signal given the transformed values.)

USAGE

N1 <HDMRD(N, LOG2N, TS)> N2

or N1 <IHDMRD(N, LOG2N, TS)> N2 for the inverse

where: N = number of samples per transform (N must be a power of 2)

LOG2N = log base 2 of N, $N = 2^{LOG2N}$

TS = time window per transform

NOTE: The actual time window used is the integer multiple of N*DT nearest TS.

OUTPUT

Each TS, this model takes N samples of its input, and calculates the transform (or the inverse transform) of these samples, producing N transformed values. The output during this time window is the N transformed values determined during the previous TS.

REFERENCE

For a discussion of this transform, see COMSAT Laboratories, "Orthogonal Transform Feasibility Study, Final Report", November 1971, Contract NAS9-11240, Section 3.3.

MODEL	GROUP ID	PAGE	DATE
HADAMARD TRANSFORM	Transform	2	April, 1972

APPLICATION

The output of the HDMRD Transform during the first TS is +1. IHDMRD uses this string of ones as a sync pulse to lock on to the transformed values.

MODEL	GROUP ID	PAGE	DATE
ORDERED HADAMARD TRANSFORM	Transforms	1	April, 1972

LIBRARY MODEL NAMES

OHMRD ORDERED HADAMARD	IOHMRD INVERSE ORDERED HADAMARD

DESCRIPTION

This model samples the input signal and outputs the Ordered Hadamard Transform of these samples. (IOHMRD - Inverse Ordered Hadamard Transform - reconstructs the signal given the transformed values.

USAGE

N1 <OHMRD(N, LOG2N, TS)> N2

or

N1 <IOHMRD(N, LOG2N, TS)> N2 for the inverse

where: N = number of samples per transform (N must be a power of 2)

LOG2N = log base 2 of N, $N = 2^{LOG2N}$

TS = time window per transform

NOTE: The actual time window used is the integer multiple of N*DT nearest TS.

OUTPUT

Each TS, this model takes N samples of its input and calculates the transform (or the inverse transform) of these samples, producing N transformed values. The output during this time window is the N transformed values determined during the previous TS.

APPLICATION

The output of the OHMRD Transform during the first TS is +1. IOHMRD uses this string of ones as a sync pulse to lock on to the transformed values.

MODEL	GROUP ID	PAGE	DATE
ANALOG-TO-DIGITAL CONVERTER	Converter	1	April, 1972

LIBRARY MODEL NAMES

ATOD

DESCRIPTION

The A/D Converter model determines a bit sequence based upon the analog input level at sampling time. The output is a binary bit-stream (0 or +1). Flexibility is provided by inputting the number of bits per word, peak input level, minimum input level, and bit time.

USAGE

N1 <ATOD(NBIT, FLOOR, CEILING, BT)> N2

where: NBIT = number of digital bits per analog word

FLOOR = a lower bound of the analog input

CEILING = an upper bound of the analog input

BT = bit time (the actual bit time is the multiple of DT closest to BT)

DETAILED DESCRIPTION

ATOD represents an analog value with a string of NBIT binary bits (0 or +1). This string of bits may be considered a binary number, B, whose value is within the range 0 (all 0 bits) to $2^{NBIT}-1$ (all one bits). FLOOR is presented by the value 0, and CEILING is represented by the value 2^{NBIT}. Each word time (NBIT*BT), ATOD constructs this number B such that the expression

$$FLOOR + B * \left((CEILING-FLOOR)/2^{NBIT}\right)$$

is as close to the analog input as the range and resolution allow.

MODEL	GROUP ID	PAGE	DATE
ANALOG-TO-DIGITAL CONVERTER	Converter	2	April, 1972

OUTPUT

Each BT, ATOD selects the next most significant bit of the digital word, to track the value of the analog input.

APPLICATION

ATOD may be used with either DTOA or SHDTOA as the decoder.

MODEL	GROUP ID	PAGE	DATE
DIGITAL-TO-ANALOG CONVERTER	Converter	1	April, 1972

LIBRARY MODEL NAMES

DTOA

DESCRIPTION

The D/A Converter determines an analog value from a binary bit stream (0 or +1) and the model parameters. This model assumes the output is continuous and attempts to produce output with as few abrupt changes as possible.

N1 < DTOA(NBIT, FLOOR, CEILING, BT) > N2

where: NBIT = number of digital bits per analog word

FLOOR = a lower bound of the analog output

CEILING = an upper bound of the analog output

BT = bit time (the actual bit time is the multiple of DT closest to BT)

NOTE: These parameters are related to the A/D Converter which coded the signal.

OUTPUT

Each bit time, DTOA examines the new bit of the digital word. If, by some combination of remaining bits, the current analog output value can be represented, then the output remains the same. Otherwise, the output is adjusted up or down until it is within range. For a description of the relationship between the input bit stream and the analog value, see Analog to Digital Converter.

APPLICATION

Since this model tries to maintain the analog output at a constant level, only changing it when it has to, this model is best for use where the analog signal is continuous.

SYSTEMS ASSOCIATES

MODEL	GROUP ID	PAGE	DATE
SAMPLE-HOLD DIGITAL-TO-ANALOG CONVERTER	Converter	1	April, 1972

LIBRARY MODEL NAMES

SHDTOA

DESCRIPTION

The Sample-hold DIA Converter determines an analog value from a binary bit stream and the model parameters. This model determines a value during one word time and outputs that value during the entire next word time.

USAGE

N1 < SHDTOA(NBIT, FLOOR, CEILING, BT) > N2

where: NBIT = number of digital bits per analog word

FLOOR = a lower bound of the analog output

CEILING = an upper bound of the analog output

BT = bit time (the actual bit time is the multiple of DT closest to BT)

NOTE: These parameters are related to the A/D converter which coded the signal.

OUTPUT

Each digital word time (NBIT*BT), SHDTOA forms an analog value from the input and its parameters (for a description of the relationship between the input bit stream and the analog value, see Analog to Digital Converter). During this word time, its output is the analog value calculated during the previous word time.

APPLICATION

This model is for use when discrete values are being decoded, such as output from the orthogonal transform models.

MODEL	GROUP ID	PAGE	DATE
MULTI LEVEL PCM	Coders	1	April, 1972

LIBRARY MODEL NAMES

MULTI LEVEL PCM
MLTPCM

DESCRIPTION

This code modulator produces an m-level signal based upon a serial input bit stream (polar or binary).

USAGE

N1 <MLTPCM(BT,M)> N2

where: BT = bit time

M = number of levels (symbols)

N = LOG_2M

The signal at the input mode (e.g., N1) is assumed to be a polar or binary bit stream.

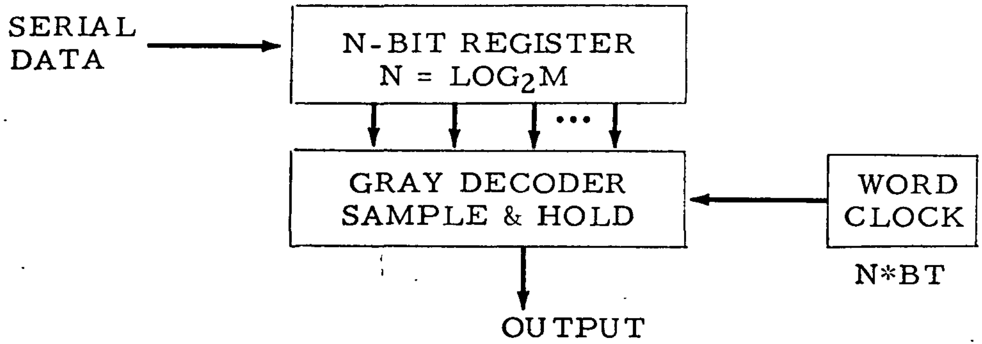
OUTPUT

A serial-to-parallel conversion is made and a gray code level selection is used in generating an output bit stream at a rate of N*BT. Output levels are normalized to a peak value of unity (+1), represented by equal levels separated by 1/(M-1). The minimum level (0) corresponds to the null symbol.

BLOCK DIAGRAM

Functionally, the model is represented as follows:

MODEL	GROUP ID	PAGE	DATE
MULTI LEVEL PCM	Coders	2	April, 1972

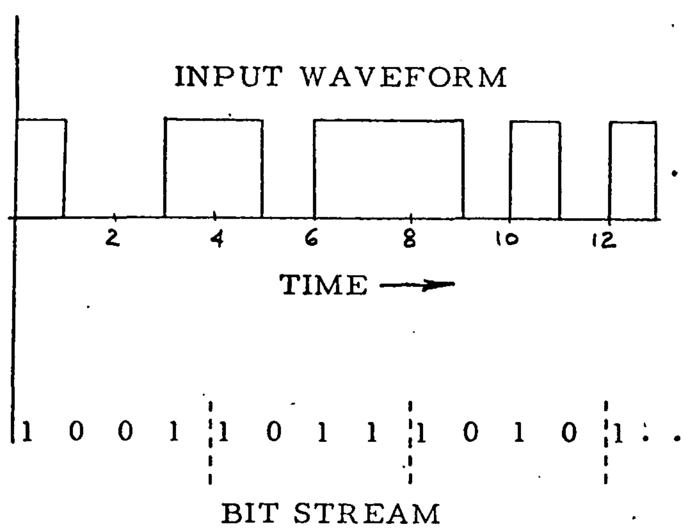


DETALED DESCRIPTION (Example)

The serial bit stream is loaded into an N-bit register. At time intervals of N*BT, the register is sampled and the output waveform value (0 to +1) is determined from the reflected (gray) code in the register. The output is held at this level for N*BT, at which time the register is sampled for the next word. The following diagram depicts the time sequence for an arbitrary input bit stream. The parameters for this example are:

M = 16 (4 Bits)

BT = 1



Register History

Time	Contents 1234	Output
0	1---	0
1	10--	0
2	100-	0
3	1001	0
4	1---	14/15
5	10--	14/15
6	101-	14/15
7	1011	14/15
8	1---	13/15
9	10--	13/15
10	101-	13/15
11	1010	13/15
12	1---	12/15
13	10--	12/15
14	100-	12/15
15	1001	14/15
16	0---	14/15
17	00--	14/15
18	001-	14/15
19	0011	14/15
20		2/15

MODEL	GROUP ID	PAGE	DATE
MULTI LEVEL PCM	Coders	3	April, 1972

APPLICATION

The multi-level coder may be used to drive the FM and PM modulators to produce m-ary PM carrier signals. Attention should be made to appropriate scaling of the MLTPCM output to produce the correct modulating signal magnitudes.

MODEL	GROUP ID	PAGE	DATE
COMPLEX ADDER	MATH	1	April, 1972

LIBRARY MODEL NAMES

COMPLEX ADDER
CADD

DESCRIPTION

The complex input to this model is added to a specified complex number passed through the argument list.

USAGE

N1 < CADD(XREAL, XIMAGE) > N2

where: XREAL = the real part of the addend
XIMAGE = the imaginary part of the addend

OUTPUT

The complex output at N2 is the complex sum of (XREAL + i*XIMAGE) and the input at N1.

MODEL	GROUP ID	PAGE	DATE
COMPLEX MULTIPLIER	MATH	1	April, 1972

LIBRARY MODEL NAMES

COMPLEX MULTIPLIER
CMULT

DESCRIPTION

The complex input to this model is multiplied by a specified complex number passed through the argument list.

USAGE

N1 < CMULT(XREAL, XIMAGE) > N2

where: XREAL = the real part of the multiplier
XIMAGE = the imaginary part of the multiplier

OUTPUT

The complex output at N2 is the complex product of (XREAL + i*XIMAGE) times the input at N1.

MODEL	GROUP ID	PAGE	DATE
DIFFERENTIATOR	MATH	1	April, 1972

LIBRARY MODEL NAMES

DIFFERENTIATOR
DIFFER
DIF

DESCRIPTION

This model differentiates the input signal utilizing the bilinear z-transform of s.

USAGE

N1 < DIFFER > NZ

OUTPUT

Let the signal at N1 at time T be INP(T).
Let the signal at N2 at time T be OUT(T).

Then:

$$OUT(T) = \left[2*(INP(T)-INP(T-DT)/DT\right] - OUT(T-DT)$$

where $OUT\ (TSTART-DT) = 0$

With this scheme, the output may alternate wildly above and below the actual differential. The fluctuation is proportional to the derivative error at time = 0, since it is initialized to zero. If the output of the differentiator is fed into a filter, this fluctuation does not matter. This scheme has the advantage that if the output of the differentiator is fed into one of the system integrators, the input samples at N1 can be reconstructed exactly.

MODEL	GROUP ID	PAGE	DATE
INTEGRAL WITH INITIAL CONDITIONS	MATH	1	April, 1972

LIBRARY MODEL NAMES

INTEGRAL WITH INITIAL CONDITIONS
INTGIC

DESCRIPTION

This model provides for integration with initial conditions.

USAGE

N1 < INTGIC(FV) > N2

where: FV = initial value of the integral at TIME=TSTART

OUTPUT

This model integrates the input at N1 using the trapezoidal rule approximation. The output at N2 is this integral plus FV, the first value (initial value) of the integral.

MODEL	GROUP ID	PAGE	DATE
INTEGRATOR	MATH	1	April, 1972

LIBRARY MODEL NAMES

INTEGRATOR
INTGRT

DESCRIPTION

This model integrates the input signal using the trapezoidal rule, which is the bilinear z-transform of 1/s.

USAGE

N1 < INTGRT > NZ

OUTPUT

The output at N2 is the trapezoidal approximation of the input signal at N1.

MODEL	GROUP ID	PAGE	DATE
DE-INTERLEAVER	MISC.	1	April, 1972

LIBRARY MODEL NAMES

DELEAV
DEINTERLEAVE

DESCRIPTION

The De-interleaver decoder real signals output by the Interleaver back into their complex form. It causes a phase delay of one sample time.

USAGE

N1 <DELEAV(N, TW)> N2

where: N = number of samples per TW

TW = time window

NOTE: These parameters should be the same as those in the Interleaver

OUTPUT

Each sample time (TW/N) this model samples the input for the real value. One half sample time later it gets the imaginary value from the input. The output of the model during this sample time is the complex value found in this way during the previous sample time.

APPLICATION

This model is used in conjunction with the Interleaver. Particularly to reconstruct complex values for input to the Inverse Fourier Transform. It is started by the sync pulse from the Fourier Transform.

MODEL	GROUP ID	PAGE	DATE
DELAY	MISC.	1	April, 1972

LIBRARY MODEL NAMES

DELAY

DESCRIPTION

This model introduces a specified time delay into a real signal.

USAGE

N1 < DELAY(LAG) > N2

where LAG = an integer number of DT's to delay the signal

OUTPUT

The output at N2 at time T equals the input at N1 at time T-LAG*DT.

MODEL	GROUP ID	PAGE	DATE
INTERLEAVER	MISC.	1	April, 1972

LIBRARY MODEL NAMES

INTLEV
INTERLEAVE

DESCRIPTION

The Interleaver allows complex values to be carried over a single line. During the first half of one sample time, the output of this model is the real part of the input. During the remainder of the sample time, the output is the imaginary part of the input.

USAGE

N1 < INTLEV(N, TW) > N2

where: N = number of samples per TW

TW = Time Window

NOTE: This model is usually used in conjunction with a model such as the Fourier Transform, where N and TW would be identical to the parameters of the Fourier Transform.

OUTPUT

Each sample time (TW/N), INTLEV samples the complex input and stores the real and imaginary parts. It them outputs the real part of the input for (TW/N)/2 and the imaginary part for the remainder of the sample time.

APPLICATION

This model was designed to appear after the Fourier Transform, or other models with complex outputs. It converts a complex valued signal to a real signal for use by other models which can only deal with real signals. See also DE-INTERLEAVER.

MODEL	GROUP ID	PAGE	DATE
PHASE SHIFTER	MISC.	1	April, 1972

LIBRARY MODEL NAMES

PHASE SHIFTER
PHSHFT

DESCRIPTION

This model causes a phase shift (delay) of a specified number of degrees.

USAGE

N1 < PHSHFT(DGREES, FC) > NZ

where DGREES = number of degrees to delay signal

FC = frequency of input signal

MODEL	GROUP ID	PAGE	DATE
SPLIT	MISC.	1	April, 1972

LIBRARY MODEL NAMES

SPLIT

DESCRIPTION

The input signal is split into its real and imaginary parts.

USAGE

N1 < SPLIT > N2

OUTPUT

The real part of N2 is set to COS (N1). The imaginary part of N2 is set to SIN (N1).

MODEL	GROUP ID	PAGE	DATE
TIME LATCH	MISC.	1	April, 1972

LIBRARY MODEL NAMES

CALINB

DESCRIPTION

This model provides a time dependent latching function which sets the output equal to the input for time LAG*DT.

USAGE

N1 < CALINB (LAG) > N2

where LAG = an integer number of DT's before the signal can pass through this model.

OUTPUT

N2 = 0 for T LAG*DT

N2 = N1 for T LAG*DT

MODEL	GROUP ID	PAGE	DATE
ZERO CROSSING DETECTOR	MISC.	1	April, 1972

LIBRARY MODEL NAMES

ZERO CROSSING DETECTOR
ZRODET

DESCRIPTION

This model detects when the input signal becomes zero or crosses the zero reference.

USAGE

N1 < ZRODET > NZ

OUTPUT

N2 is generally set to zero. If, during one simulation step (DT) the signal at N1 goes to zero or crosses zero, N2 is set to 1 for that DT only.